UTT-110B Series VOIP Gateway (SIP)

User manual

VER 1.0

Copyright

This manual refers to the sole ownership by the Shenzhen Ultrative Technology Co., Ltd. (hereinafter referred to as Ultrative) exclusive ownership without the written permission of the Company, any unit or individual is not entitled to any form of copy, disseminate and reproduction in part or all of the contents of this manual, Ultrative reserves the right to charge against law.

Note: This manual is for the user of UTT-110B Series VOIP gateways; the company has the final interpretation of the manual and might improve, at any time, the products mentioned and the manual itself without prior noticing.

This document contains proprietary intellectual property rights owned by Shenzhen Ultrative Technology Co., Ltd, all rights reserved.

Unless mentioned elsewhere, Shenzhen Ultrative Technology Co., Ltd., is the only registered trademark in China and worldwide. For more information, please contact us directly

Windows 98/NT/2000/XP/2003/7 Windows operating systems mentioned in this manual, its trademarks and products are copyright owned by Microsoft Corporation.

Company: Shenzhen Ultrative Technologies Co.,Ltd.
Address: 011#, 6/F, Block A, Hedong Bldg., Liutang Rd.,
Xixiang St., Bao An District, 518102, Shenzhen, China
Hotline: +86(755) 29630367
Postcode: 518102
Webpae: http://www.Ultrative.com.cn
Email : support@Ultrative.com

INDEX

UTT-110B SERIES VOIP GATEWAY (SIP) 1

COPYRIGHT 2

PART I MANUAL GUIDANCE 5

1.1 PURPOSE 5
1.2 TARGET READERS 5
1.3 ABOUT THE CONTENT 5
1.4 REMARKS 5

PART II PRODUCT INTRODUCTION 6

2.1 CHARACTERISTICS OF UTT-110B SERIES VOIP GATEWAY 6
2.2 UTT-110B SERIES SPECIFICATIONS 8
2.3 UTT-110B MODEL NAME 8
2.4 PACKAGING 8
2.5 APPEARANCE 9
2.5.1 Products Panel Diagram 9
2.5.2 LED Indicators 9
2.6 HARDWARE CONNECTION 10
2.6.1 Connection to LAN by Static IP or DHCP 10
2.6.2 As a proxy server is responsible for dial-up Internet 10
2.7 NETWORK ACCESS CONFIGURATION 11
2.8 LOG-IN TO THE WEB CONFIGURATION INTERFACE 11

PART III BASIC WEB SETTINGS 14

3.1 SYSTEM MANAGEMENT 14
3.2 NETWORK CONFIGURATION 16
3.2.1 WAN Settings 16
3.2.2 LAN Setting 18
3.2.3 Route Setting 19
3.3 SIP SETTINGS 20
3.4 CALLPATH 22
3.4.1 Add a call path 22
3.4.2 Add a call Rule 23
3.5 PORT SETTINGS 24
3.5.1 Port Basic Settings 25
3.5.2 Advance Settings 26
3.6 PHONE NUMBER SETTING 28
3.6.1 Single port phone number setting 28
3.6.2 Port bulk configuration 29
3.7 SYSTEM TOOL 29
3.8 PROGRESS TONE CONFIGURATION 30
3.9 SYSTEM STATUS 31

PART 4 IVR INQUIRY AND IP ADDRESS CONFIGURATION 32

4.1 WAN PORT IP INQUIRY AND CONFIGURATION 32
4.2 LAN PORT IP INQUIRY AND CONFIGURATION 33
4.3 INQUIRY PHONE NUMBER OF THE PORT 33

PART FIVE TYPICAL APPLICATION CONFIGURATION(16FXS+16FXO) 33

5.1 CONFIGURATION OF FXS+FXO PORT EQUIPMENT FOR DIAL "9" IN SECONDARY DIAL 33
5.2 FXS+FXO EQUIPMENT FXO PORT CONFIGURATION— CORRESPONDING FXS PORT 38

Part I Manual Guidance

1.1 Purpose

In order to help users of our devices to understand and use our UTT-110B Series GWs more effectively, hereby, we present this User Manual with our sincerity. This Manual consists of all detailed information that one need to know about the products.

1.2 Target Readers

The target readers of this manual includes:

I.T. Engineer

Sales Engineer

NOC

1.3 About the content

UTT-110BSeries VOIP Gateway (SIP) User manual offers detailed hardware specifications, installations, allocation and LED indication, together with elaborate WEB configuration demonstration.

This manual has the following:
Part I: Manual Guidance
Part II: Product Introduction
Part III: Basic WEB Settings
Part IV: IVR Inquiry, IP address setting.
Part V: Typical Scenario

1.4 Remarks

All the following Examples are based on ***UTT-7500-16FXS16FXO*** as the only subject.

Part II Product Introduction

UTT-110B Series VOIP Gateway gives way to standard IP Audio/Fax/Data services, which is also called Integrated Access Device or Access Gateway. Normally, in a NGN, UTT-110B belongs to the Access Layer of the network. Its main role is to combine all network terminals into a unified web, in order to make all services possible in the network. By adhering all traditional circuit exchange features, UTT-110B further delivers advantages that IP technology can bring, making smooth migration from traditional PSTN to NGN possible; At the meantime, UTT-110B can deliver value-add services just within the traditional PSTN network, providing a more flexible and balanced choice for customers. This series—UTT-110B VoIP Gateways, supports 1-2 channels of VoIP communications, has been widely used in Government Agencies, Commercial Organizations and Large Corporate for their own communication network. It is an ideal product to be used in where VoIP communication is required.

2.1 Characteristics of UTT-110B Series VOIP Gateway

Carrier-class reliability

Support Efforts to improve fault detection, network alarm functions.

Low Power Consumption and High Density integration.

Supports 3rd Level lightning protection

POTS Interfaces support over-current protection.

Using ripple smaller, higher-quality communication power, support surges, power lines and other protective lap, output stability, high reliability, and supports instantaneous power protection.

Using electromagnetic radiation shielding properties of the chassis, electromagnetic compatibility, ROHS and so do the professional design, can effectively shield electromagnetic interference variety of environments.

Transmission loss, loss frequency, nonlinear distortion, crosstalk attenuation, noise, and non-cross-cross heavy heavy noise and other indicators have reached the telecommunicationsstandard.

Flexible, powerful security policy

Support administrator login and password protection, built-in firewall function, can effectively prevent the various network virus attacks, and improve data security.

Multiple protocol support capabilities

Support the SIP protocol.

Support SNMP network management protocol, centralized network management devices, remote monitoring and maintenance.

Support T.30, T.38 voice pass-through protocol, fax service on IP bearer network.

Support RTP / RTCP protocol, to achieve real-time voice packet encapsulation and voice playback.

Audio Services support capabilities

Support for voice, fax, Modem services.

Support a variety of basic voice services and value-added services.

IP telephony and traditional PSTN phone switch.

Flexible access

Support IP line access.

Support xDSL dial-up access.

Support Cable Modem access.

Diversity management

Support for SNMP-based remote centralized network management device.

Web-based network management support equipment.

Powerful QoS guarantee

Based on IPv4 Tos and DiffServ support services to ensure the voice priority.

Support IEEE802.1P, IEEE802.1Q.

Multi-adjustable parameters

Including the supply voltage can be adjusted, the loop current, ringing voltage, long-term, short-term, impedance parameters and so on.

Advanced voice processing technology

Support ITU-T G.711a/mu, G.729, G.723.1, and other speech coding.

Support voice activity detection (VAD), effectively save network bandwidth resources.

Support Comfort Noise Generation (CNG).

Support echo cancellation, up to 64ms.

Supports adaptive dynamic buffering technique.

Supports packet loss compensation.

Support DTMF generation / detection.

Support Caller ID detection and display functions.

Support DTMF band, SIPINFO, RFC2833 transmission technology.

Support flexible input / output gain control.

1:1 Lifeline function supported.

Support one phone dual-number function

2.2 UTT-110B Series Specifications

Graph 2-1 UTT-110B Series Specifications

Project	UTT-110B Series
Adaptor (Input / Output)	Input: 100-240V Output: 12V 1A
Interface (WAN)	10/100Base- T RJ-45 for LAN, Auto MDIX
Interface (LAN)	10/100Base- T RJ-45 for PC, Auto MDIX
Power Consumption	Idle: 4 W / full load: 6W
Operating Temperature	-5 ~ 50 ℃
Relative Humidity	5 ~ 95% non-condensing
The main chip	5VT-1310
DSP	5VT-1310
CODEC	ZL88601
Flash	32 MB
SDRAM	256MB
Dimensions (Lx H x W)	116mm × 91mm × 30mm
Weight	140g

2.3 UTT-110B Model Name

Graph 2-2 UTT-110B Model Name

Product Name	FXS	FXO
UTT-110B-1FXS	1	0
UTT-110B-2FXS	2	0
UTT-110B-1FXS1FXO	1	1

2.4 Packaging

Before installing, make sure that the product packing list:

UTT-110B Gateway *1

Power Cord *1

Product Manuals CD * 1

Product warranty card *1

Network cable *1

Telephone lines * 1-2

2.5 Appearance

2.5.1 Products Panel Diagram

Image 2-1 Front Panel

Image 2-2 Rear Panel

2.5.2 LED Indicators

Table 2-2 Front panel connectors and LEDs

Front panel connectors and LEDs	Description
Alarm	Alarm LED, all the ports open registration, while not registered as a flashing softswitch, softswitch registration Alarm goes off.
Active	Status indicators, the normal operation of the lamp is flashing.
Power	Power indicator, turn the lights connected to the power supply for the long bright state.
1-2	Port work light, off-hook, ringing, the lights are flashing during a call, the standby is off.

Table 2-3 Rear panel connectors and LEDs

The rear panel connectors and LEDs	Description
ON / OFF	Power switch, ON / Off
AC 100-240V	Power cord interface, connect the power cord.
WAN	Equipment upstream interface, when in the 10M Ethernet port rate, the green light, orange light off; When working in the 100M Ethernet port speed, green and orange lights are on, when the flow of data out of date, the green light, orange lights flashing.
LAN	Device configuration interface, when in the 10M Ethernet port rate, the green light, orange light off; When working in the 100M Ethernet port speed, green and orange lights are on, when the flow

	of data out of date, the green light, orange lights flashing.
1	FXS connected to a telephone or PBX trunk interfaces
2	FXO interfaces connected to the PSTN or PBX extension

2.6 Hardware Connection

2.6.1 Connection to LAN by Static IP or DHCP

1) applies has internal LAN or home users.
2) WAN port UTT-110B Series integrated access devices connected to the hub or switch, as shown in Figure 2-3.
3) WAN port based on the local area network environment, using PPPoE dial-up mode, dynamically obtain IP (DHCP) or static IP mode.

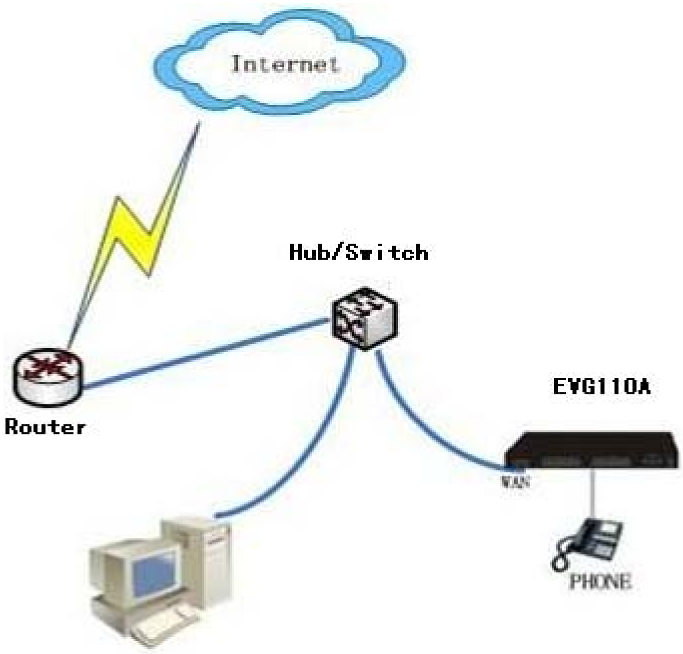


Figure 2-3 Series integrated access devices in the LAN connection

2.6.2 As a proxy server is responsible for dial-up Internet

1) UTT-110B Series Voice over IP Integrated Access Device Modem WAN port directly connected with ADSL (Cable), as shown in Figure 2-4.
2) UTT-110B Series Voice over IP Integrated Access Device as a proxy server, agent in charge of the Internet.

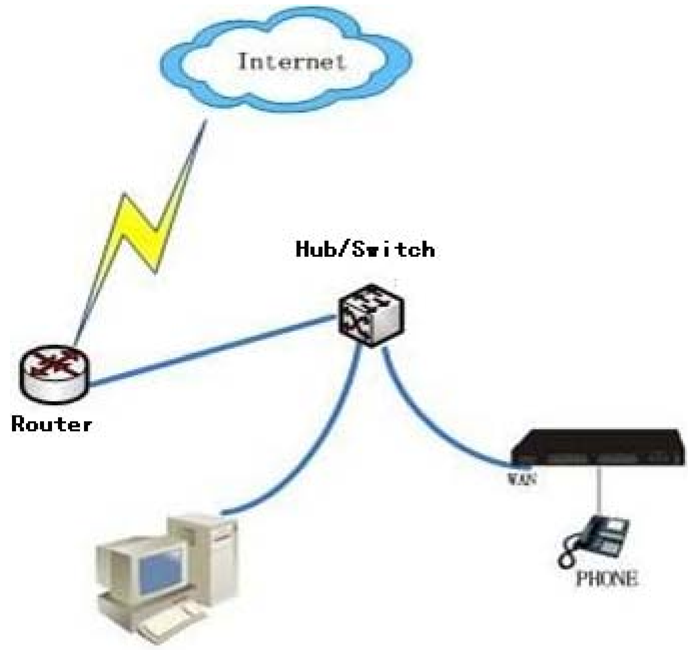


Figure 2-4 UTT-110B Series Integrated Access Device as a proxy server connection

2.7 Network Access Configuration

Firstly confirm the connection: WAN port UTT-110B Series integrated access devices support PPPoE, dynamic IP address or a static IP address mode

2.8 Log-in to the WEB Configuration Interface

1) selection has a computer card and TCP / IP protocol installed, the computer and UTT-110B Series Voice over IP integrated access device's LAN port to connect to a hub or switch with a network cable, network cable can also be used to connect directly to the computer and the LAN port.

2) Turn on the computer "My Network Places" and "local connection", right click and choose Properties. Below, the IP address of the computer with UTT-110B Series Voice over IP integrated access device's LAN port IP address is configured on the same network segment. (UTT-110B Series The factory default LAN port IP voice integrated access device is IP is**192.168.11.1,** subnet mask is **255.255.255.0.)**

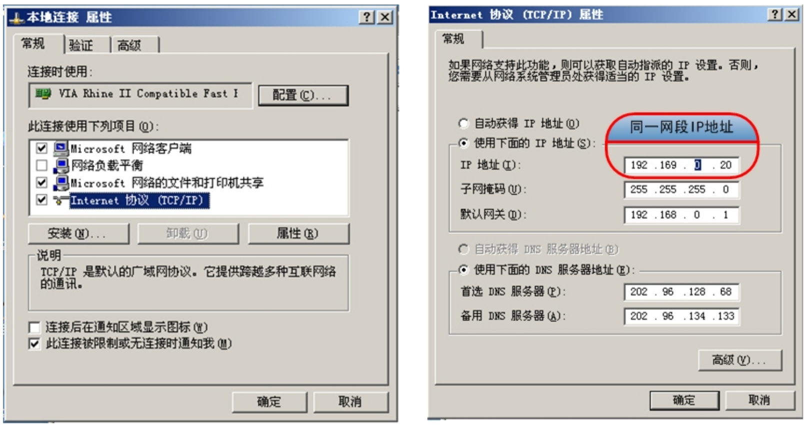


Figure2-5 PC Ipaddress Setting

Ping command to test whether and UTT-110B Series IP voice integrated access devices connected properly.

C: \> ping 192.168.11.1

Pinging 192.168.11.1 with 32 bytes of data:

Reply from 192.168.11.1: bytes = 32 time <1ms TTL = 255

Reply from 192.168.11.1: bytes = 32 time <1ms TTL = 255

If the above prompt appears, indicating that the computer has access to the normal communication can be integrated Voice over IP devices and UTT-110B the series.

C: \> ping 192.168.11.1

Pinging 192.168.11.1 with 32 bytes of data:

Request timed out.

Request timed out.

If the above message appears, it means that the computer and the UTT Series Voice over IP integrated access devices connected nowhere please first check your UTT-110B Series Voice over IP integrated access device is connected properly (under normal circumstances, AN port status LEDs are point bright), and then enter the "Internet Protocol (TCP / IP) Properties" page to see if your computer's IP address is configured correctly.

3） Click  ,input at the address bar **http://192.168.11.1** (LAN Default IP: **192.168.11.1**), then:

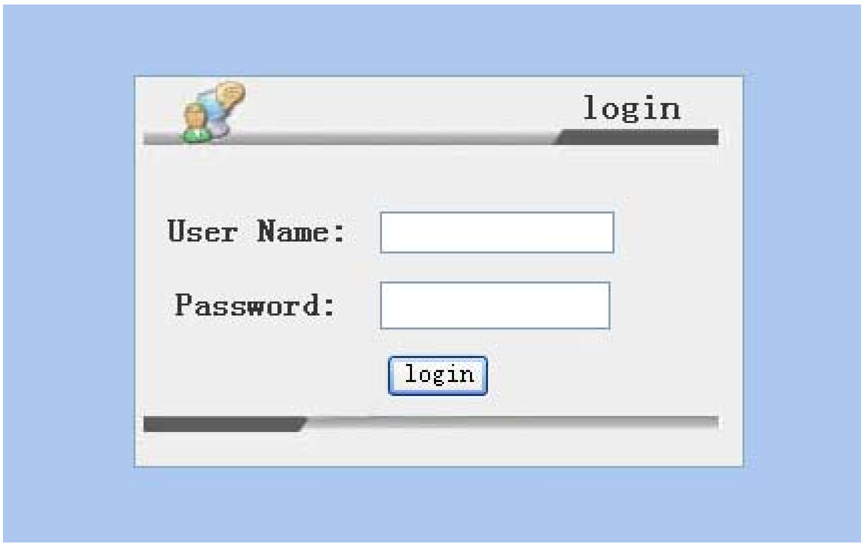


Figure2-6　WEB Login Interface

User Name:**admin**,Password: **admin**, （Default Username: **admin** passowrd: **admin**）,Click Enter then,

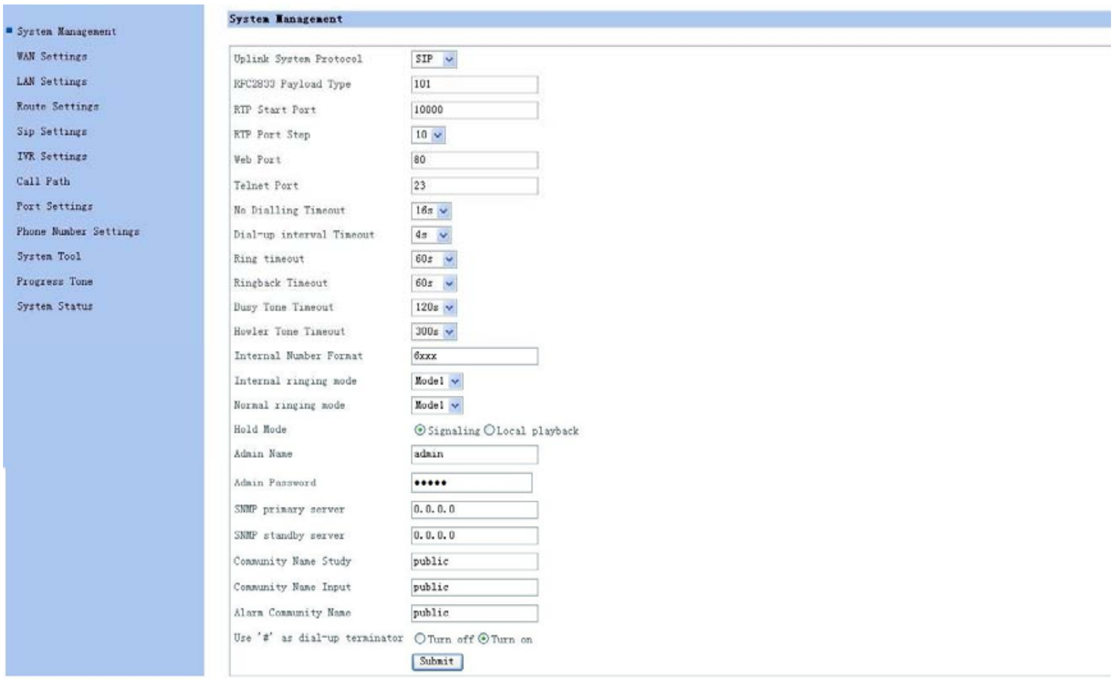


Figure 2-4　WEB Setting interface

Part III Basic WEB Settings

3.1 System Management

Login Succeeded, then go to System Management

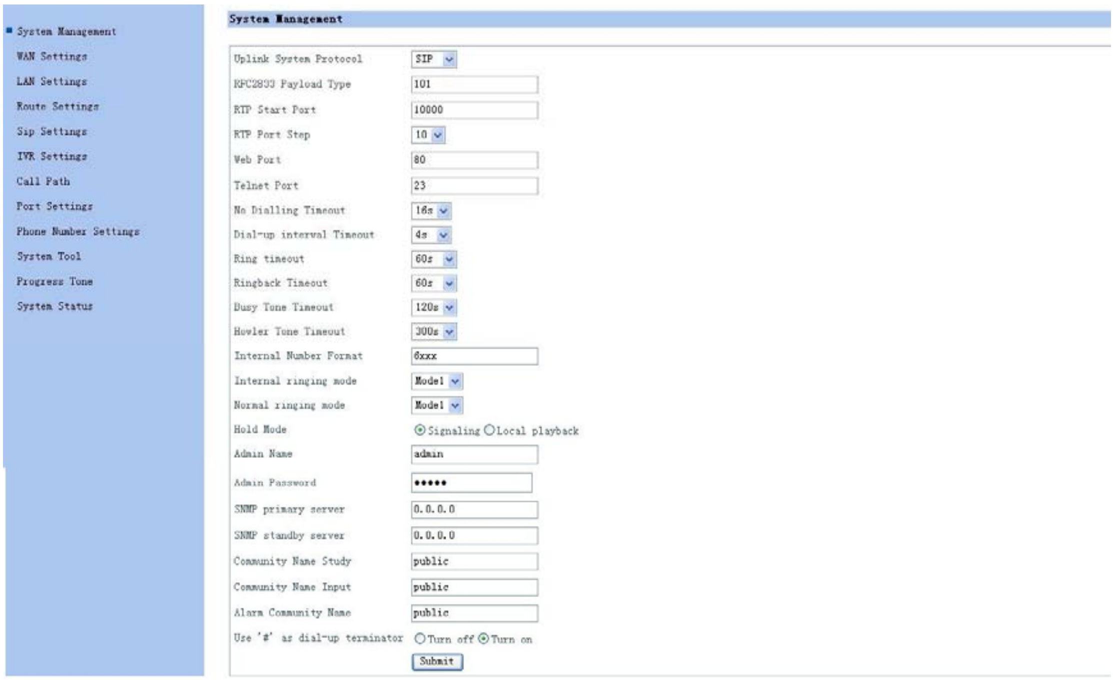


Figure3-1 System Management Interface

Graph 3-1 System Management

System Management Configuration Item	Descriptions
System upstream agreement	Default device using SIP protocol dropdown MGCP/H248 temporarily not take effect.
RFC2833 payload type	With DTMF mode "rfc2833" use. Default value 101, a limited range of values from 97 to 101. Default, the parameters to be consistent with the peer device, but it can also auto-negotiation.
RTP start port	Min sending and receiving RTP port, this parameter can not be less than 3000, it is recommended to configure the default value can not be less than 10,000, and can be modified.
RTP port step	RTP step parameter settings, the default port 10, the drop-down can be modified.
WEB port	Log WEB configuration interface of the portfacilities, with a default value of 80, can be modified.
Telnet port	telnet port used to configure the device, the default

	23, can be modified.
Hook without dialing timeout	The default value is 16s, the drop-down can be modified.
Between dialing timeout	The default value 4s, drop-down can be modified.
Ringing Timeout	Telephone ringing timeout, the default 60s, the drop-down can be modified.
Ringback Timeout	Hear the ringback tone timeout, the default 60s, the drop-down can be modified.
Busy Timeout	Hear a busy tone timeout, the default 120s, drop-down can be modified.
Howler tone timeout	Hear the sound of howler timeout defaults 300s, drop-down can be modified.
Extension number format	Line with the distinction between inside and outside the ring to use Caller ID defaults 6xxx format for extension number, use the intercom ringing pattern.
Inside the ring pattern	Mode 1:1 S pass 4S off; Mode 2:2 S pass 4S off; model 3:0.5 S pass through 0.5S 5S 4S broken off; Mode 4:1 S can be modified throughthe drop-down 3S off.
Normal ringing mode	Mode 1:1S pass 4S off; Mode 2:2S pass 4S off; model 3:0.5S pass 0.5S off 0.5S pass 4S off; Mode 4:1 S can be modified throughthe drop-down 3S off.
Hold mode	The default value for the signaling mode, you can select local playback.
Administrator name	Default administrator name admin, can be modified.
Administrator Password	Default admin password admin, can be modified.
SNMP master server	Fill in this SNMP master server IP address or domain name.
SNMP standby server	In this alternate fill SNMP server IP address or domain name.
Read community name	Fill in this read community name.
Write community name	Fill in this write community name.
Alarm group name	Fill in this alarm group name.
Use the '#' sign as a dial terminator	After opening, '#' key to dial a terminator, after the retreat, '#' key to send a number to call.

3.2 Network Configuration

3.2.1 WAN Settings

Click “Wan Settings” to modify configurations

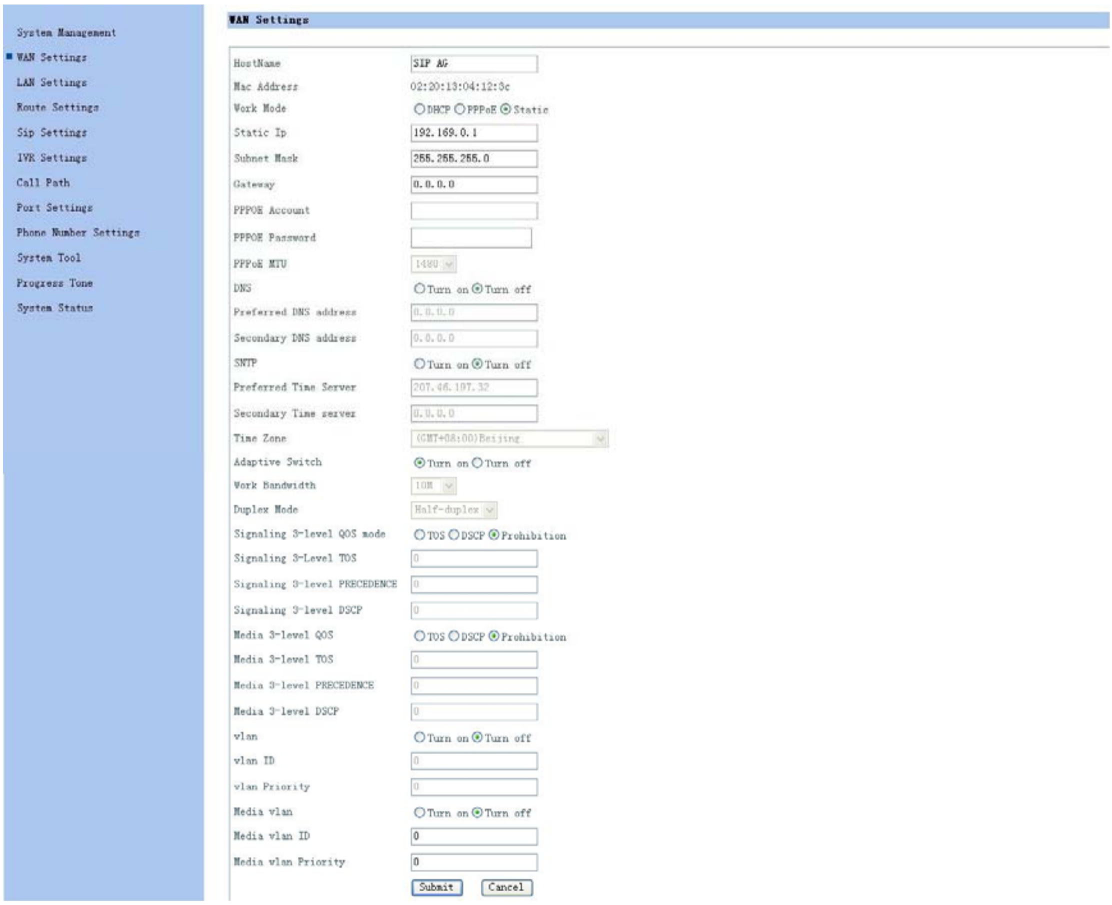


Figure3-2 WAN Setting Interface

Table 3-2 WAN configuration

WAN configuration items	Description
Host name	Names can configure the device, the device defaults to the hostname SIP AG, the user name of the device can be configured as required.
MAC address	Display the MAC address of the WAN port.
Operating Mode	WAN port mode DHCP: Open the DHCP mode, using dynamic host configuration protocol to obtain an IP address and other network parameters; PPPoE: PPPoE open mode; Static: fixed IP mode.

Static IP address	When the operating mode using When "Static", the correct input on the configuration items available IP address, the device defaults IP address: 192.169.0.1.
Static Mask	Mask with the IP addresses, operating mode using When "Static", you must configure the mask, the default value Mask: 255.255.255.0.
Static Gateway	LAN device where the gateway IP address, operating mode using When "Static", you must configure the gateway address, the default value of the static gateway: 0.0.0.0.
PPPoE account	When the operating mode using PPPoE mode, enter the correct PPPoE account available, no default value.
PPPoE password	When the operating mode using PPPoE mode, enter the correct password PPPoE available, no default value.
PPPoE MTU	When the operating mode using PPPoE mode, PPPoE MTU default value 1480, the drop-down can be modified.
DNS switch	DNS service is off by default, when you need to enable select Open.
Preferred DNS address	DNS server is turned on, the preferred DNS address Default: 0.0.0.0, this can be modified in the preferred DNS server address.
Secondary DNS address	DNS server is turned on, the default secondary DNS addresses: 0.0.0.0, this can be modified alternate DNS server address.
SNTP switch	SNTP service defaults to off, select open when you need to enable.
Preferred Time Server	Preferred time server IP address Default: 207.46.197.32, this can be modified in the preferred time server IP address.
Standby time servers	Standby time server IP address Default: 0.0.0.0, this can be modified spare time server IP address.
Time Table	Select the time zone, Default: time zone (GMT +08:00) Beijing, pull-down can be modified.
Adaptive Switch	The default value is an adaptive switch is turned on, when you close the WAN port can be configured manually operating speed and duplex mode.
Work rate	Adaptive switch to select off, choose to work in this rate WAN port.
Duplex mode	Adaptive switch to select off, in this selection WAN port duplex mode.
Signalling three QOS mode	The default value is disabled, choose TOS or DSCP.
Signalling three TOS	QOS signaling mode is selected as three-TOS, the TOS three signaling default value is 0, the effective range of values 0 ~ 7. IP precedence 6 and 7 are used for network control

	communications use, is not recommended to use.
Signalling three PRECEDENCE	When three QOS signaling mode is selected as TOS,signaling three PRECEDENC default value of 0, the effective range of values 0 ~ 7. IP precedence 6 and 7 for network control communications use is not recommended to use.
Signalling three DSCP	When the signaling mode is selected as three QOSDSCP, signaling three DSCP default value is 0.
Media three QOS mode	The default value is disabled, choose TOS or DSCP.
Media three QOS	Media TOS three QOS mode selection when signaling three TOS default value is 0, the effective range of values 0 ~ 7. IP precedence 6 and 7 for network control communications use, is not recommended to use.
Media triple PRECEDENCE	Media QOS mode is selected as three-TOS, the signaling three PRECEDENC default value is 0, the effective range of values 0 ~ 7. IP precedence 6 and 7 are used for network control communications use, is not recommended to use.
Media three DSCP	When the media three QOS mode is selected as DSCP,signaling three DSCP default value is 0.
VLAN	The default value is off, select Open when needed.
VLAN ID	After VLAN open, VLAN ID default value is 0, the effective range of 1 to 4094
VLAN Priority	After VLAN open, VLAN priority default is 0.
Media VLAN	The default value is off, you need need to open.
Media VLAN ID	After the media VLAN enabled, the media VLAN ID default value is 0, the effective range of 1 to 4094.
Media VLAN Priority	After the media VLAN enabled, the media VLAN priority default is 0.

3.2.2 LAN Setting

Click “LAN Settings” to modify configurations

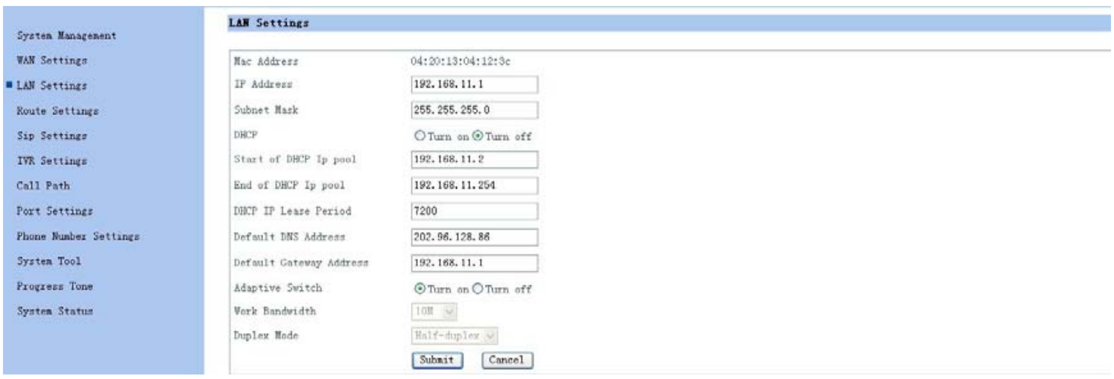


Figure3-3 LAN Setting

Table 3-3 LAN configuration

LAN configuration items	Description
MAC address	Display the MAC address of the LAN port.
IP addresses	LAN IP address of the default is: 192.168.11.1, change the IP address in this LAN port.
Mask	LAN port mask defaults: 255.255.255.0, in this modification LAN port mask.
DHCP	DHCP server defaults to off, select Open when needed.
IP pool starting address	After the DHCP server is turned on, connected to the LAN port to obtain an IP network terminal starts from that address.
End IP address pool	After the DHCP server is turned on, connected to the LAN port of the network terminal in front of the address to obtain IP.
Lease Term	IP address lease duration, the default value 7200.
The default DNS address	Default DNS Address: 202.96.128.86.
The default gateway address	Default Gateway address: 192.168.11.1.
Adaptive switch	The default value is an adaptive switch is turned on, turned off manually configure the LAN port speed and duplex mode of work.
Work rate	Adaptive switch to select off, choose to work in this rate LAN port.
Duplex mode	Adaptive switch to select off, in this select LAN port duplex mode.

3.2.3 Route Setting

Click “Route Settings” to modify configurations

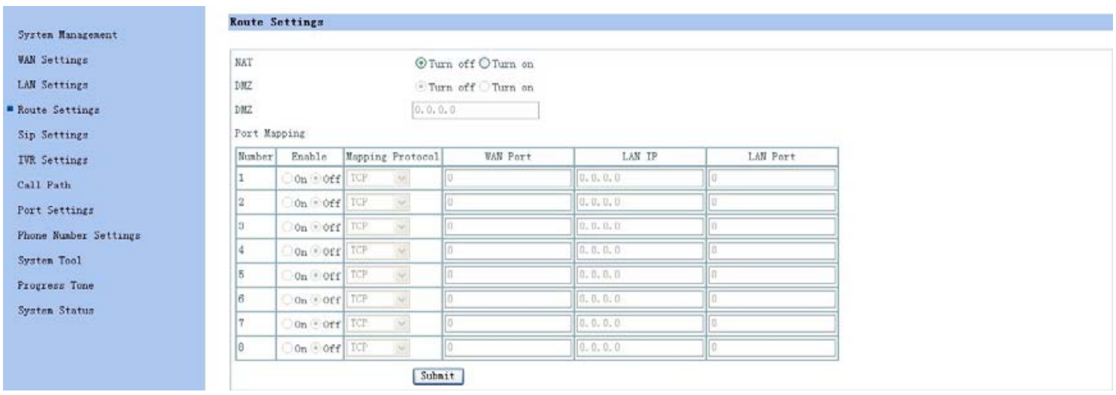


Figure3-4Route Setting Interface

Table 3-4 routing configuration

Routing configuration items	**Description**
NAT is enabled	When closed, the device retreat route forwarding; When turned on, the device opens the route forwarding.
DMZ	When NAT function is enabled, DMZ feature defaults to off, select Open when needed.
DMZ server address	In the NAT feature is turned on, when the DMZ function is turned on, fill in this DMZ address.
Port Mapping	In the NAT feature is turned on, turned off when the DMZ, port mapping is enabled by default turned off, turned on, drop down to select the mapping protocol mapping fill WAN port, LAN mapped address, LAN port mapping these parameters.

3.3 SIP Settings

Click Sip Settings to modify in the interface below

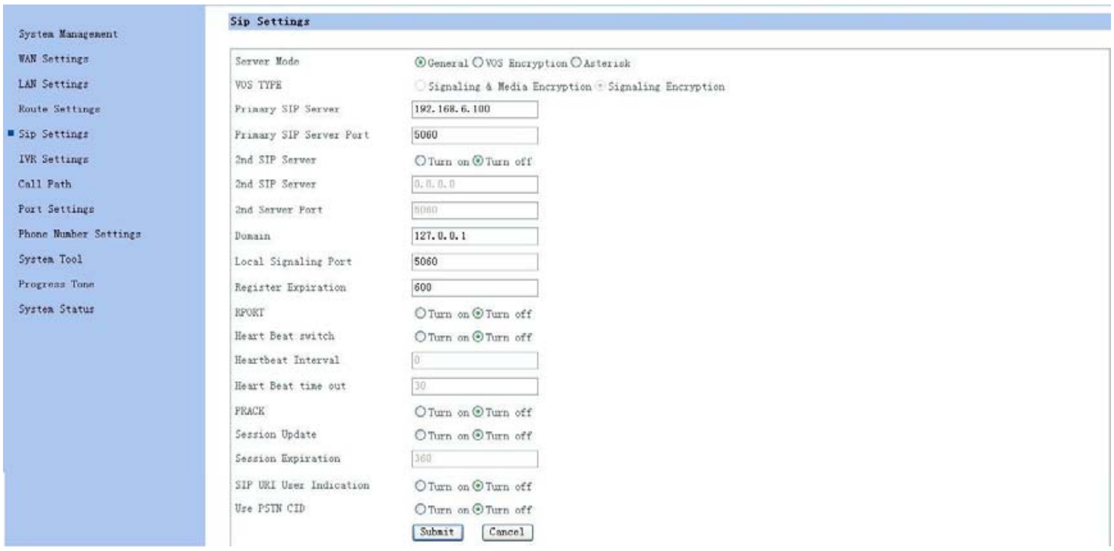


Figure3-5 SIP Setting Interface

Table 3-5SIP configuration

SIP configuration items	**Description**
Server mode	General: All mainstream standard SIP protocol server; VOS Encryption: Encryption exchange for VOS made soft;

	Asterisk: When soft switch as Asterisk, optional for this option.
VOS type	When the server mode selection VOS encryption, media encryption or choose optional signaling encryption.
The primary server	Configure the primary SIP server's IP address or domain name.
Primary server port	Configure the primary server SIP register port, the default value 5060.
Backup server switch	Standby server switch, the default is off, you need to use an alternate server is turned on.
Standby server	After the standby server opens, enter the backup SIP server's IP address or domain name.
Standby server port	After opening the backup server, enter the alternate port SIP registration server.
Domain	Fill sip server's domain, under normal circumstances, and fill the same SIP server IP address for the domain; butt IMS, IMS platform to fill in the domain name.
Local signaling port	This equipment SIP signaling port, the default value 5060.
Registration refresh time	SIP registration refresh time, in seconds, the default value 600, the actual registration refresh time and softswitch negotiation.
rport	The default value is off, select Open with rport required field.
Heartbeat switch	Heartbeat off, do not send a heartbeat message to the SIP server; heart open, it will send option heartbeat information to the SIP server.
Heartbeat interval	Send heartbeat interval, the default value of 0 seconds, can be modified.
Heartbeat Timeout	Send heartbeat timeout, during which time the scope of this SIP server if the response has not been the heartbeat information that has been disconnected from the server, the default value of 30 seconds, and can be modified.
PRACK	When turned on, invite support 100rel; closed, invitedoes not support 100rel
Session Update	When turned on, support UPDATE; when closed, does not support UPDATE
Session Update	After the update session is open, the session update the default value is 360, the actual conversation time updates and softswitch negotiation.
SIP URI parameter to carry User	When open, SIP URI will carry user = phone parameter; When closed, SIP URI does not carry user = phone parameter.
Use PSTN CID	This parameter is only for the FXO port to the PSTN get to use the explicit when turned on, to display the number of

	PSTN number; closed to display the number of FXO port number.

3.4 CallPath

In WEB setting interface, Select" Call Path", users can see a default "DigitMap-Default", besides this default Call Path, user can manually add his own. So far we support maximum 4 different call paths. Each call path can set different rules in order to control its authorities,(Local Calls, Long Distance Calls, Oversea Calls). When the call path is set, users can go to "Port Settings"→"Basic Settings" to choose the one to activate, as shown below:

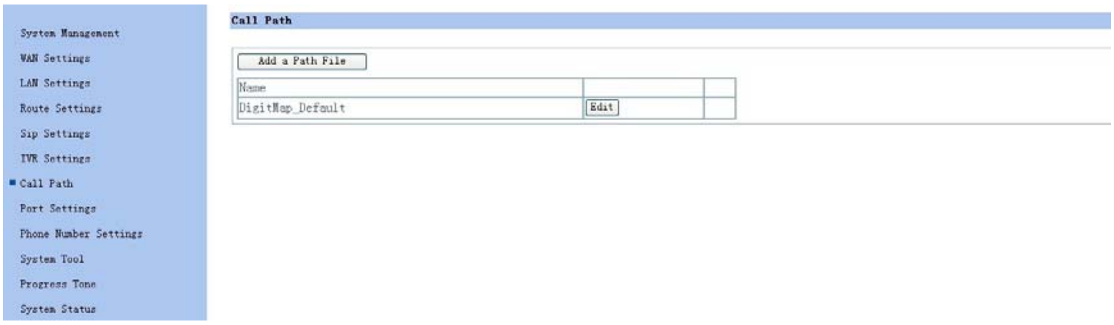


Figure3-7 Call Path Setting Interface

By Clicking "Edit", User can check the detailed rules, call rules are consisted by "0-9, .,*,#,X"(represents number 0-9,)and []" For example, [1,3,4-6,9]=13,4,5,6,9.

When Routing IP is 0.0.0.0, then the call will be sent to the server address edited in "SIP Settings; If the routing IP is a specified IP then the calls will be forwarded accordingly(p2p), as below.

3.4.1 Add a call path

Click in the bar "Path File Name",fill in the right information,"Submit"then job done, multiple call rule adding supported. As show in the interface below

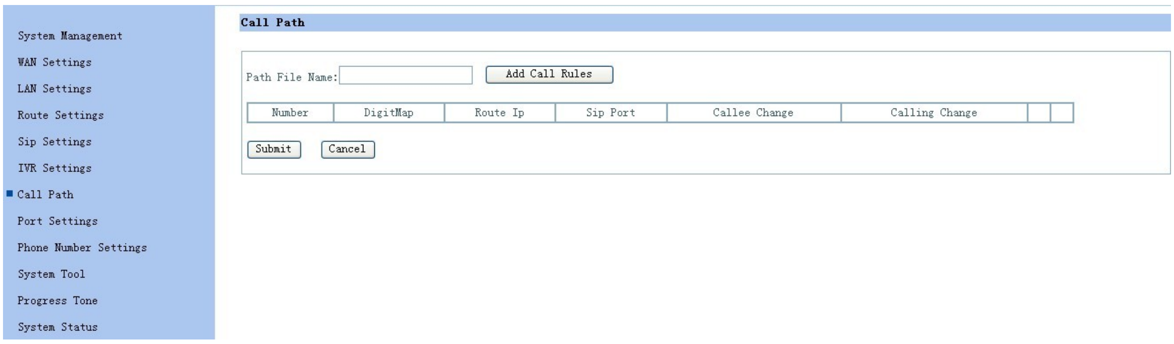


Figure3-7 Add a call path interface

3.4.2 Add a call Rule

In the call path of configuration interface, click the "Edit" option, enter the path to the file configuration interface, the following Figure:

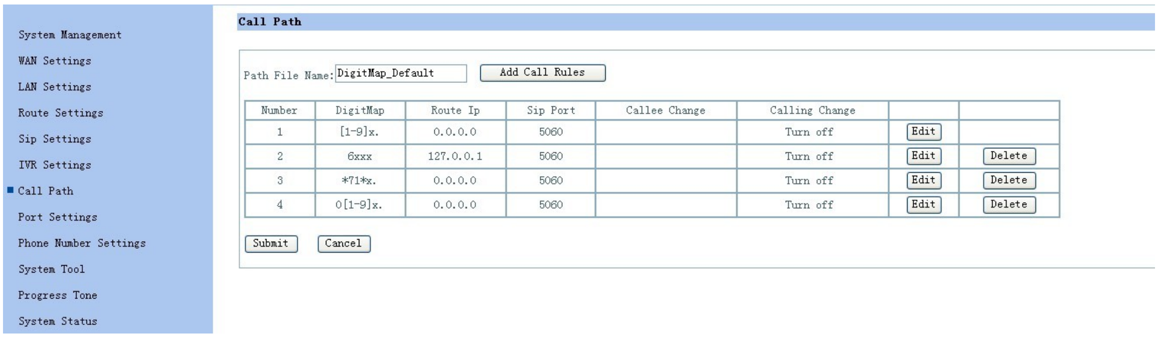


Figure3-8 Edit call path interface

Default value is DigitMap_Default, There are 4 call rules within, as shown in the above figure. Click “add call rules” then the interface goes to below

Figure3-9Add call rule interface

Table 3-7 Call rule configuration increases

Increased call rule configuration items	Explanation
DigitMap	Called number matching rules.
IP Routing	The purpose of routing IP address, the default value0.0.0.0, For the called number to the specified IP address,then fill in the IP address of the remote device.
Signaling port	The purpose of routing IP signaling port, the default value 5060.
Called number conversion	In this fill the called number conversion rules.
Caller ID conversion	The default value is off, after opening in the "Basic Configuration" option each port inside the "Caller ID transformation" option to fill the calling number transformation rules.

No conversion example	In 6xxx call rules, for example, explain number conversion number conversion by A (addition), D (delete), C (change) change the number in three ways: A (addition): There may be an increase in the number of call rules, such as call rules 6xxx, the number is converted to (a0755) 6xxx, when a user dials 6002, after sending out a number of transform 07,556,002; (Note: when necessary, call rules A front inside any character can fill (a + p)). D (delete): You can delete the rule in which the callnumbers, such as call rules 6xxx, the number is converted to6x (d) x, it means that the third deleted when users dial the number 6002 transformed sent out after the number of 602; (Note: A number may be a need to remove the direct conversion of (d), followed by d without adding deletecontent) C (c hange): to change the rules on the inside callnumbers, such as the number dialed rules 6xxx, the number is converted to 6xx (c99), sent out after transformation rules when users dial 6002 number is 60099 (Note: when necessary,Any character can be changed by calling the rules inside(c + content)) D (delete) and C (change) represents one of the rules which call regular character, and A (addition) is added in front of the character in a certain call rules can also be combined,such as call rules 6xxx, number conversion to (a0755) 6x (d) (c99)(a111), sent out when the user dials the converted number is6002 07556099111. 6x (d) (c99) is carried out at the number of transformation 6xxx are 4 , the number of bits to be consistent.

3.5 Port Settings

Click Port Settings:

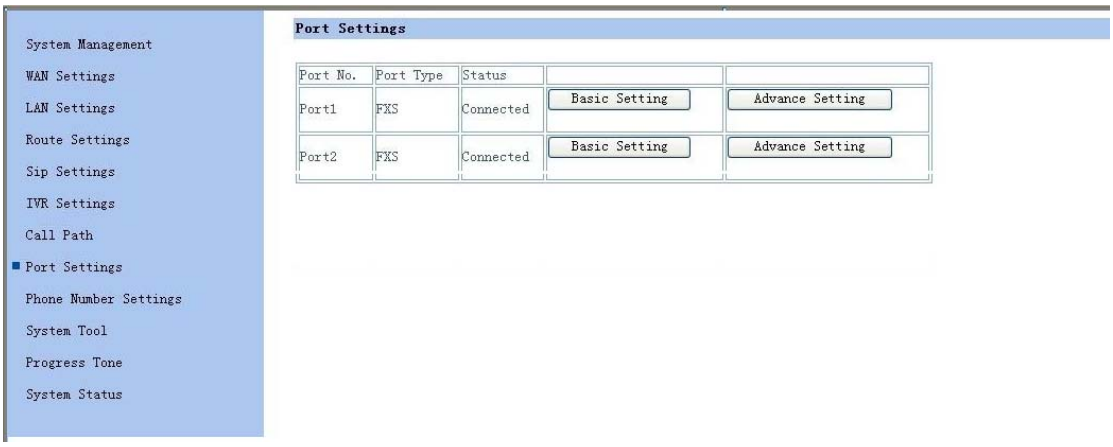


Figure3-10 Port Setting Interface

3.5.1 Port Basic Settings

Click Port Basic Settings, then WEB interface shows as below,

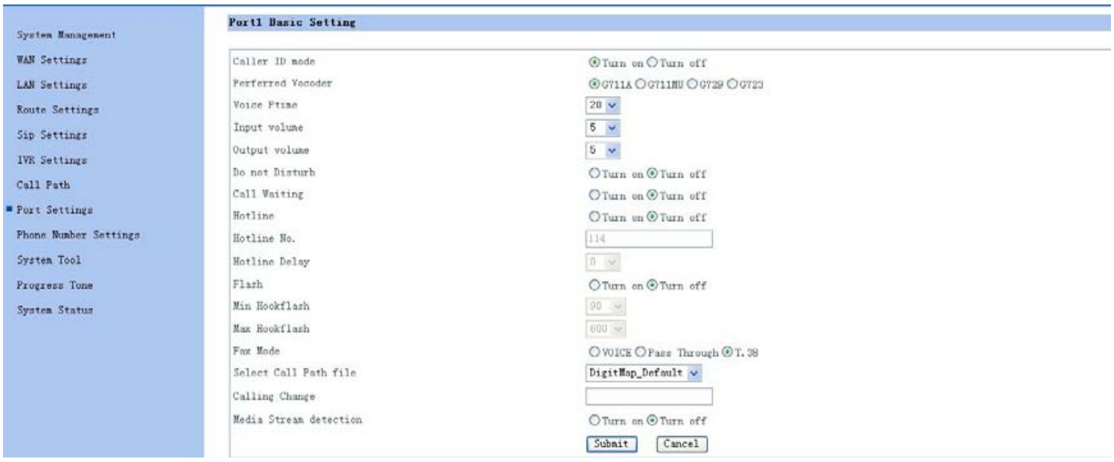


Figure3-10 Port1Basic Settings

Table 3-8 Port 1Basic Settings

Port Basic Settings	**Description**
Caller ID Mode	Default FSK, while supporting DTMF, Caller ID is not required, Can be set to off
Voice Codec Priority	Priority defaults G711A, while support is down by priorityG711U/G929/G723 so on.
Length of Audio Packagin	The default value is 20ms, support auto-negotiation.
Input volume	Set the port input volume size.
Output volume	Set the size of the port output volume.
Do Not Disturb switch	When turned on, the port open DND; closed, shut down the port Disturb feature.
Call waiting switch	When turned on, the port open call waiting function; Whenclosed, the port turn off call waiting feature.
Hotline switch	When turned on, the port opens hotline function; When closed,the port is closed hotline function.
Hotline number	After the switch is turned hotline, enter the hotlinenumber.
Hotline delay	The default value is 0 seconds for immediate hotline way; modify the default value other for the delay Hotline way.
Hook FLASH switch	When turned on, the port open for Hook Flash function; On the contrary, when turned off, all relevant functions goes off.(***Including: Call Waiting. Call Hold(RFC2543 or RFC3264) 3 way Calling, Call Transfer Unattended/Blind)***

Hook-Flash Upper limit	The default value is 90ms, support drop-down menu modification
Hook Flash Lower Limit	The default value is 600ms, support drop-down menu modification
Fax mode	The default value of T38 mode, transparent mode, or VOICEmode as necessary.
Select the call path of the file	Default is the default Call path file, if multiple call path required, then it can be modified from the drop down menu
Caller ID conversion	With the call path changeover switch inside the calling number to use, after opening, the calling number to fill in this transformation rules, transformation rules reference called number conversion rules.
Media Detection	This setting is for the FXO port, when turned on, if not detected PSTN FXO port side of the media stream, FXO port will automatically hang up when closed, is not detected.

3.5.2 Advance Settings

Click Advance Settings in Port Settings, then interface goes like the figure below:

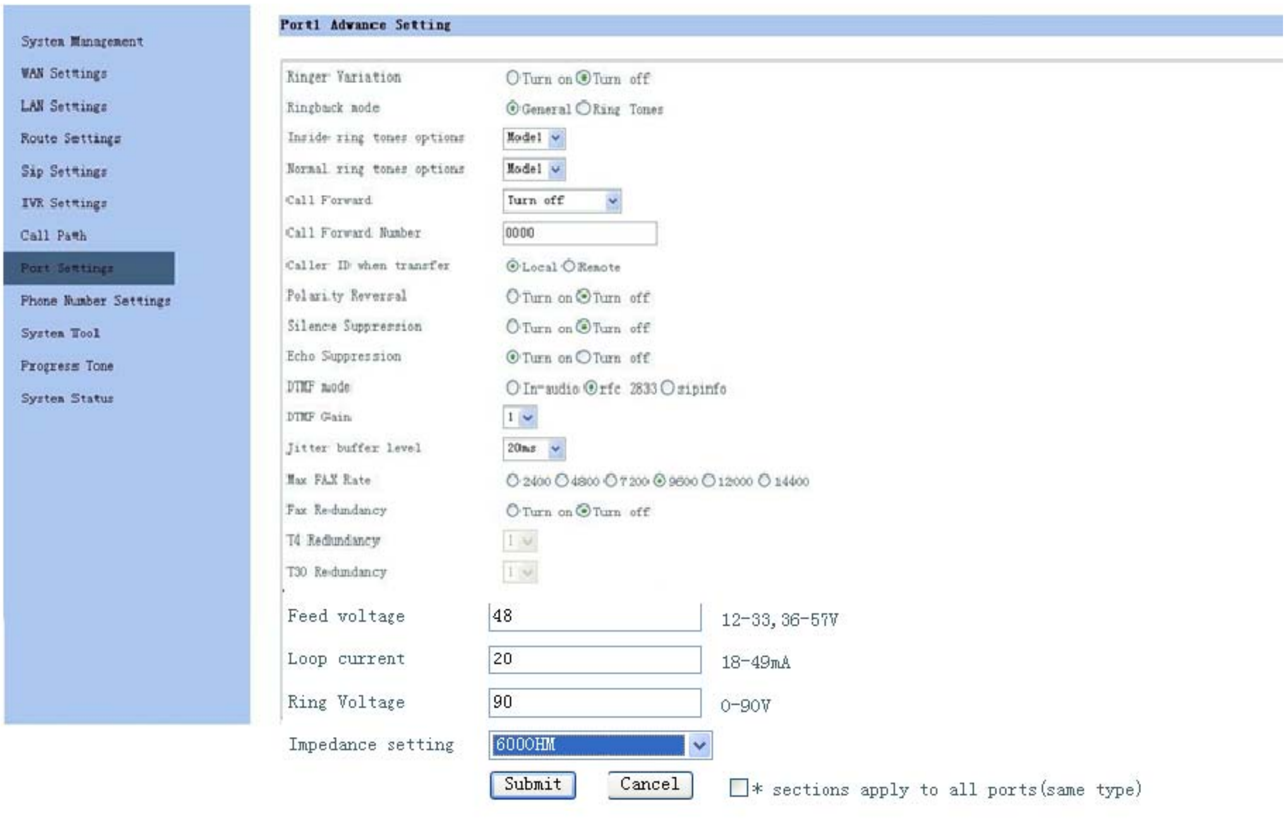


Figure3-10 Port 1 Advance Setting Interface

Table 3-9 Port 1 Advanced Configuration

Port Advanced Configuration Item	**Explanation**
Inside and outside the line ringing	When turned on, both inside and outside the port open lines ringing function; When closed, the port is closed and outside line ringing function.
Ringback way	Ordinary time, the port is common ringback way; When ringtones, ring tones open the port function.
Inside RBT mode	The default mode is a drop-down can be modified.
Normal ringing mode	The default mode is a drop-down can be modified.
Forward mode	The default is off Unconditional Forward: All the phone numbers dialed were transferred to the forwarding number; Busy turn: the number when the line is busy, dial the number for all calls are transferred to the forwarding number; Forward No Answer biography: dial the number, no one answered when this number, calls to the forwarding number.
Forwarding number	Enter the correct forwarding number available.
Forward Caller ID service	When selecting a local display local number; choose remote that displays the number of the remote number.
Reverse polarity Support	The default value is off, reverse polarity signal when turned on, the phone is turned on when the port will provide anti-polarity signal, the terminal device can use this signal telephone billing applications.
Silence Suppression	When turned on, the port opened silence suppression function; When closed, the port is closed silence suppression function.
Echo suppression	When turned on, the port open echo suppression; When closed, the port is closed echo suppression.
DTMF mode	The default is rfc2833. Band: DTMF signals along with voice transmission; rfc2833: The DTMF signal to rfc2833 format with RTP packet transmission; sipinfo: sipinfo the DTMF signals transmitted.
DTMF gain	DTMF tones to set the volume of information.
Jitter buffer level	Jitter buffer, to help overcome the effects of network jitter caused by defaults 20ms, the drop-down can be modified.
Fax maximum rate	T38 fax setting the maximum rate, the default value is 9600.
Fax redundancy	When turned on, the port open the fax redundancy; Whenclosed, the port is closed fax redundancy.
T4 redundancy	Setting the number of data packet T.38 redundant frame.
T30 redundancy	Setting the number of data packet T.30 redundant frame

Supply voltage	The default value is 48, Unit V, the effective range of the value of 12 ~ 33,36 ~ 57V.
Loop current	The default value is 20, the unit mA, the effective range of the value of 14 ~ 49mA.
Ringing voltage	The default value of 90, unit V, the effective range of0 ~ 90V.

3.6 phone number setting

Sign in web configuration interface, select “phone number setting” for setting the phone number, as the following figure(Figure 3.11)

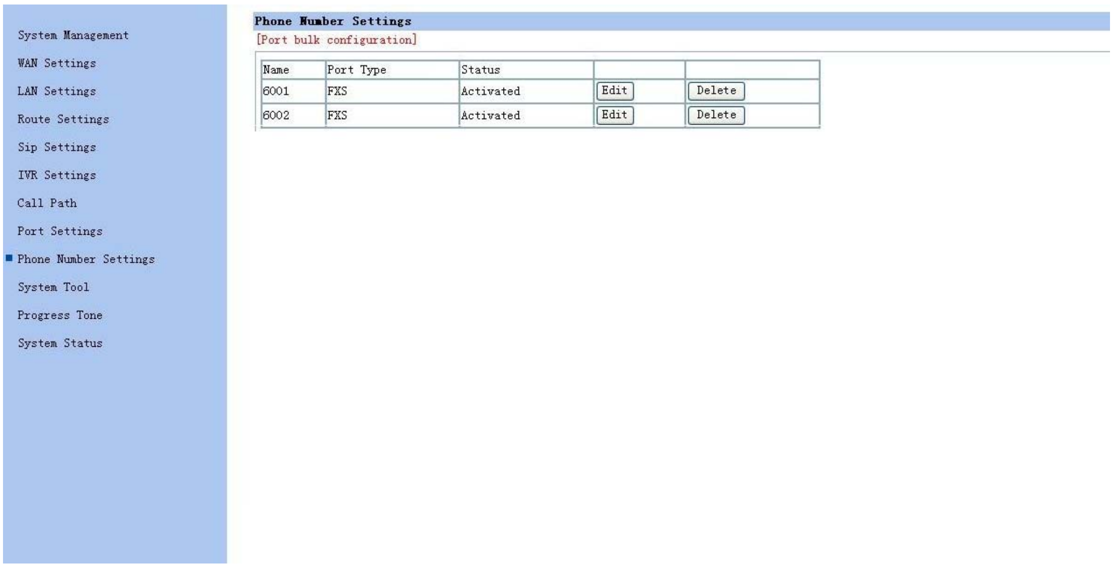


Figure 3-11 phone number setting interface

3.6.1 Single port phone number setting

Click “Edit”, enter the corresponding single port setting interface, as the following figure(Figure 3-12):

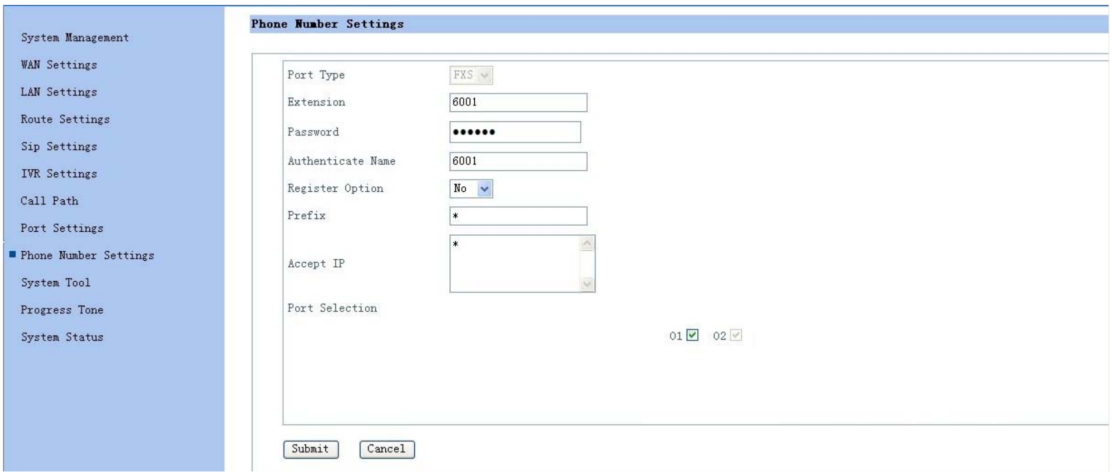


Figure 3-12 Single port phone number setting

Table 3-10 single port phone number configurations

Single port setting item	Description
Port type	The equipment automatically recognize the port is FXS port or FXO port
Phone number	Write the right phone number
Password	Write the right password
Authenticate Name	Write the right authenticate name, usually same as thephone number
Register option	Select Yes or No
Prefix	Default *，represent any prefix
Accept IP	Default *，represent any IP
Port selection	Select the port, one port number can band multiple port and realize one number multi-channel.

3.6.2 Port bulk configuration

Click “Port bulk configuration”，enter the configuration interface, writhe the phone number, password, register name, and select yes or no for register option, as the following figure(Figure 3.13):

Figure 3-13 Port bulk configuration

3.7 System tool

Sign in WEB configuration interface, select System tool, and you can see the following figure(Figure 3.14):

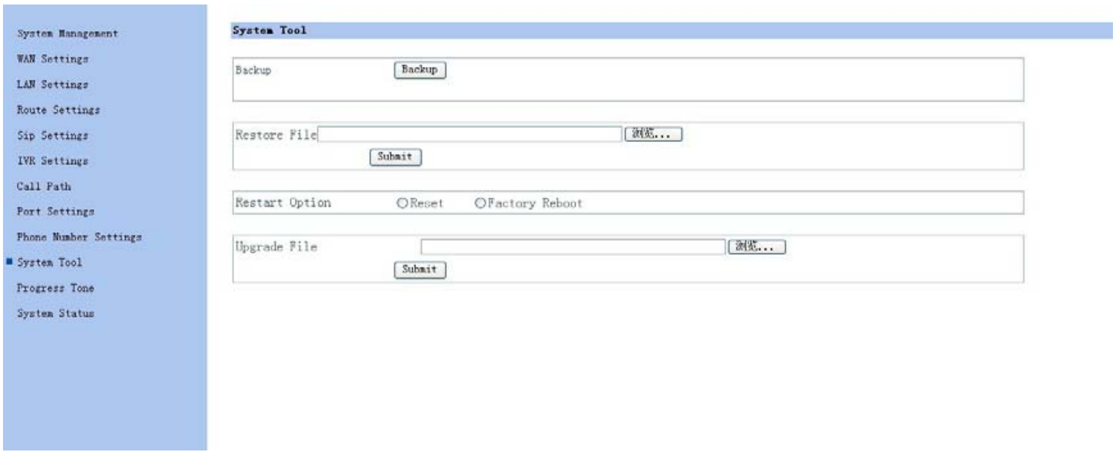


Figure 3-14 system tool configuration

Table 3-11 system tool configuration

Configuration item for system tool	**Description**
backup	Backup the configuration data for this equipment
Restore file	Select the right backup file, and restore the configuration data for this equipment
Restart option	Reset: reset the equipment Factory reboot: restore the default factory setting and restart the equipment。
Software upgrade file	Select the right software and upgrade the software version of the equipment

3.8 Progress tone configuration

Sign in web configuration interface, and select “ Progress tone” to configure the progress tone, as the following figure(Figure 3-15)

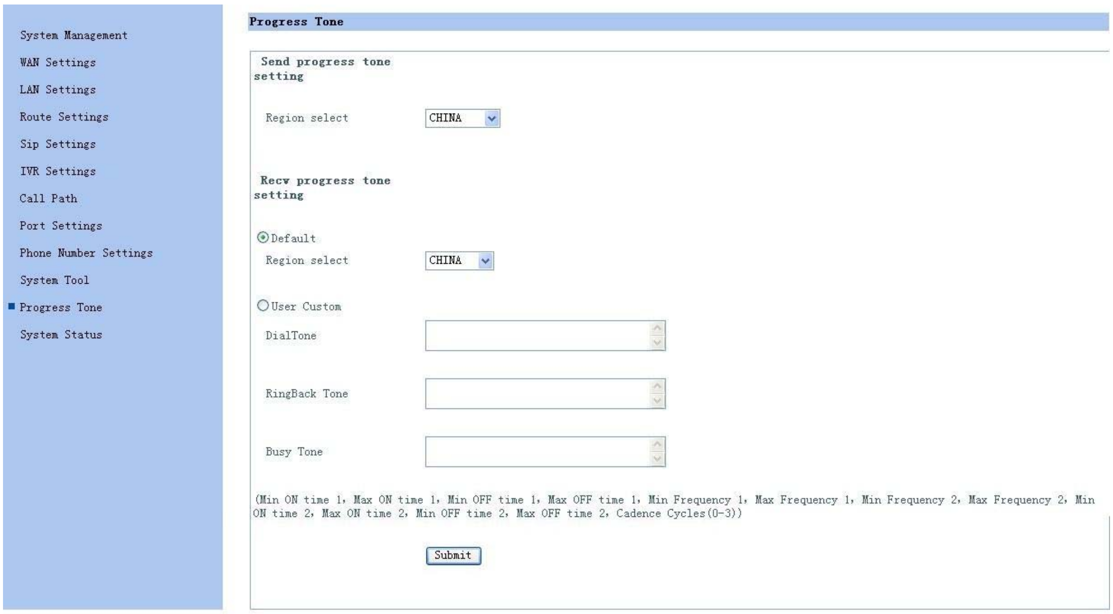


Figure 3-15 Progress tone configuration interface

Table 3-12 Progress tone configuration

Configuration item for progress tone	**description**
Send progress tone setting	Set progress tone sending by FXS port
Region select	Default value are China, can select Taiwan, Japan, Korea, United States, Germany, Russia and etc.
Progress tone receive setting	Setting the receive progress tone for FXO port
Default	Default value is China, can select China, Taiwan, Japan, South Korea, United states, Germany, Russia
User custom	Can set any of the above country/Area, including the busy- tone, ring back tone, each parameter is separate by “,” .

3.9 System status

Sign in web configuration interface, select “system status” and you can see the following Figure(Figure 3.16 System status):

Figure 3-16 System status

System status	Description
System status	Show product type, system run time, system time and system start time
Version information	Show software version and hardware version
WAN information	Show link status, MAC address, IP assignment setting, IP address, subnet mask, default gateway, preferred DNS server, 2nd DNS server
LAN information	Show connect state, MAC address, IP address, subnet mask, Start of DHCP IP pool, End of DHCP IP pool, Active DHCP clients
Route information	Show NAT start state (Turn on or Turn off)

Part 4　IVR inquiry and IP address configuration

4.1 WAN port IP inquiry and configuration

When hearing dialing tone or busy tone after phone off hook, type the following function code:

***100#　　　　(Inquiry wan port IP address)；

***101#　　　　(Inquiry wan port subnet mask)；

***102#　　　　(Inquiry wan port out gateway IP)；

***103*192*168*6*100#　　(Set WAN port IP to 192.168.6.100，or can configure the actual required IP address) ；

***104*255*255*255*0#　　(Set wan port subnet mask to 255.255.255.0，or can configure to actual required subnet mask)；

***105*192*168*6*1# (Set Wan port gateway IP as 192.168.6.1，or can configure to actual required gateway IP address)

4.2 LAN port IP inquiry and configuration

When hearing dialing tone or busy tone after phone off hook, type the following function code:

***200# (Inquiry LAN port IP address)；

***201# (Inquiry LAN port subnet mask)；

***202*192*168*10*100# (set LAN port IP to 192.168.10.100，or can configure to actual required IP address) ；

***203*255*255*255*0# (set LAN port subnet mask to 255.255.255.0，can set to actual required subnet mask)。

4.3 Inquiry phone number of the port

When hearing dialing tone or busy tone after phone off hook, type the following function code:

***300# (inquiry phone number of the port)

Note: After Setting IP address successful, it will take effect after on hook, no need restart the equipment. You can use the new IP address to sign in.

After start the immediate hotline service for the port, you cannot inquiry and configure the IP address.

Part Five Typical application configuration(16FXS+16FXO)

5.1 Configuration of FXS+FXO port equipment for dial “9” in secondary dial

Sign in Web configuration interface, and the detail configuration process is as follows:

1select “Phone numbersetting”, as following figure(Figure 5-1):

Figure 5-1 phone number configuration

2Click " Delete" Button, delete all the default number in FXO port, as following figure(Figure 5-2):

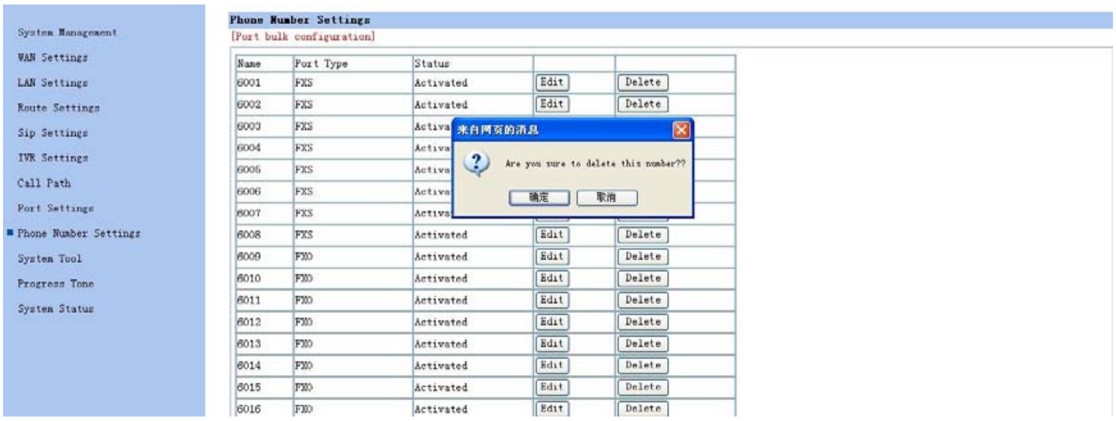


Figure 5-2 Delete phone number

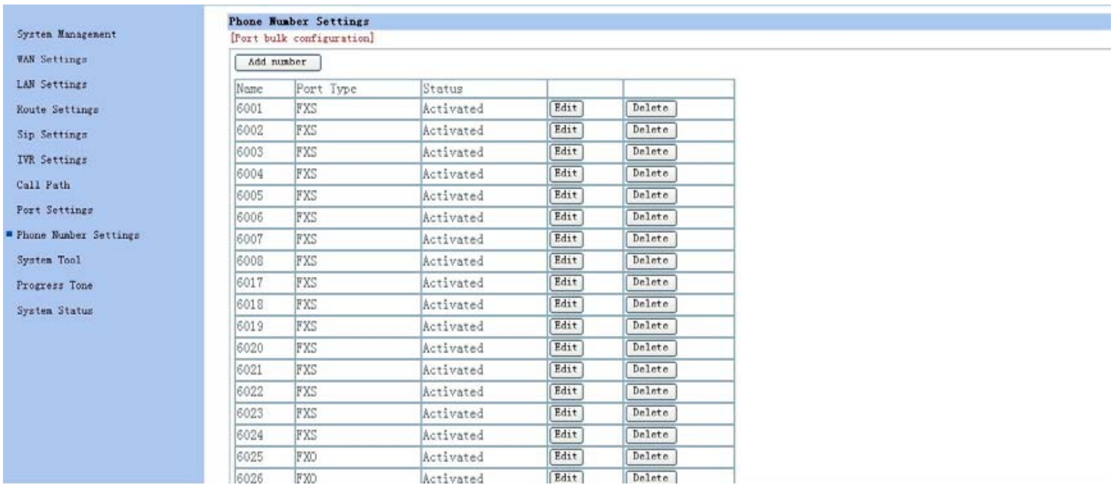


Figure 5-3Delete the phone number

3 Click "add phone number" button, add a new phone number 9, and select all the FXS port to the group of phone number 9, select "FXO" as port type, as following figure(Figure 5-4):

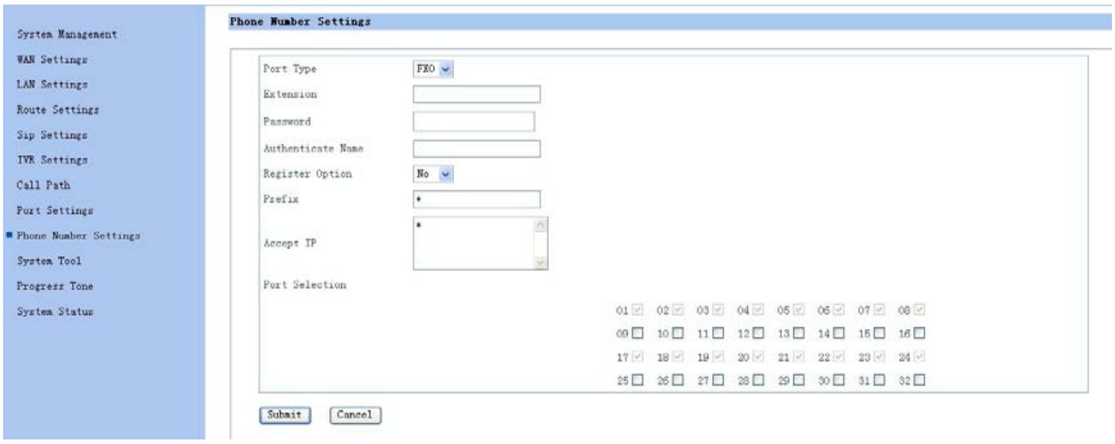


Figure 5-4 Add phone number

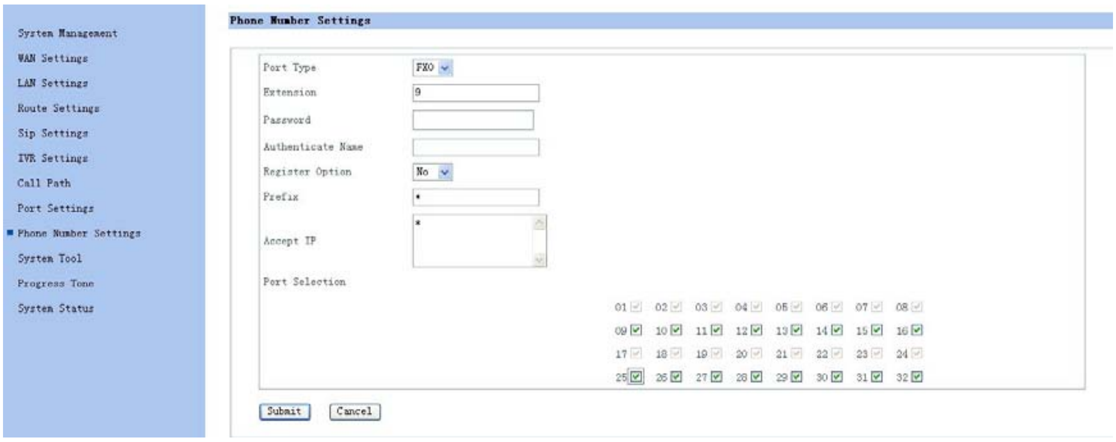


Figure 5-5 Add Phone number

4 After"submit", return to "phone number configuration" interface, and you can see the new phone number 9 with port number type FXS, as following figure(Figure 5-6):

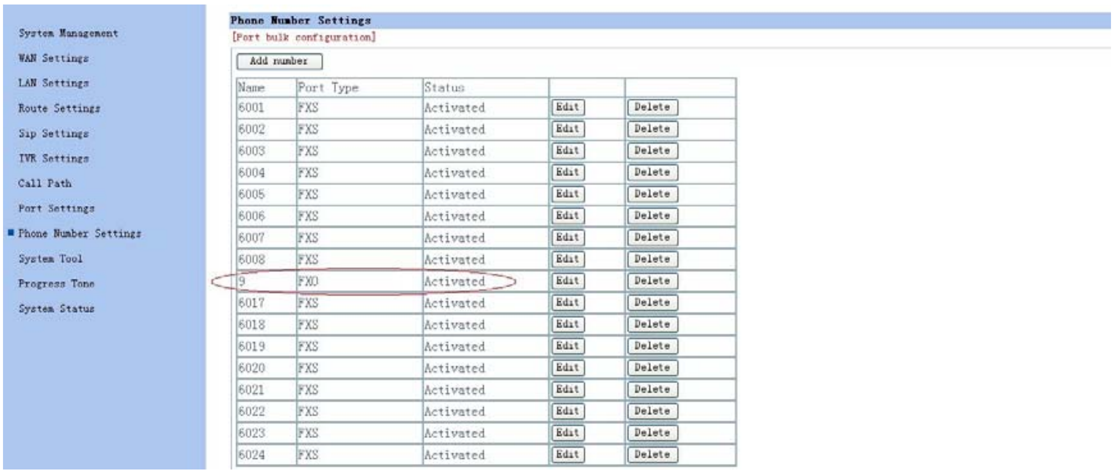


Figure 5-6 Add phone number

5 Enter "Call path" configuration interface, edit the "'Digitmap_Default'item, as following figure(Figure 5-7)

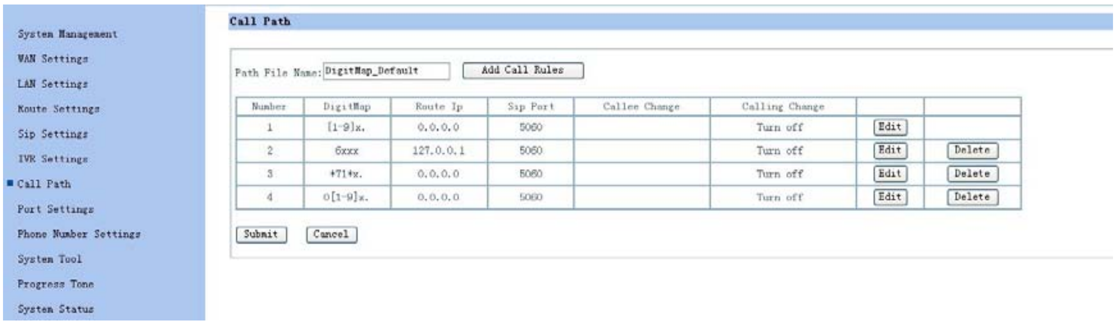


Figure 5-7 Edit call path

6Click "add call rules" button, add a call rule "9" with route IP 127.0.0.1(**Terminal loopback**)，and click " submit" button, as following figure(Figure 5-8):

Figure 5-8Edit the call path

7 Edit the default digitmap [1-9]x.，change to[1-8]x.，as following figure(Figure5-9, 5-10 and 5-11):

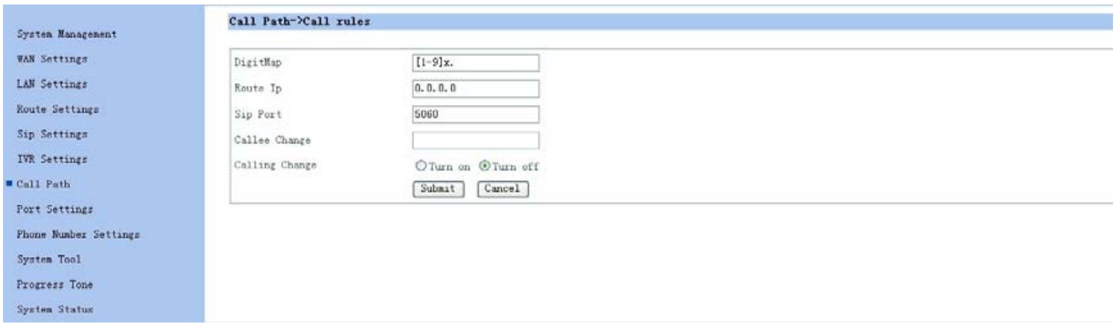


Figure 5-9 Edit call path

Figure 5-10 Edit call path

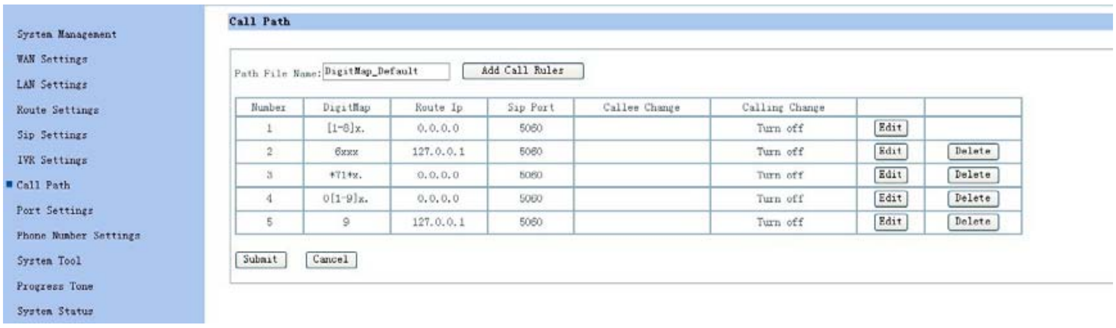


Figure 5-11Edit call path

8 After finish the above configuration, FXS port off-hook and dial 9, you can hear secondary dialing tone(the secondary dialing tone is released by the phone wire of FXO port), redial and the called number is out.

Scene summary:

This scene described the configuration method of FXS port out through FXS port for the FXS+FXO port equipment. The out number is 9, and this out number can be amended according to actual requirement. In this scene we band number 9 to all the FXO port for the equipment and form to a group for number 9. You can also set different numbers banding to different FXO ports and form to multi number group. And you can dial different number group and go out throughdifferent FXO port.

The Expanded application for this scene: If the FXS port which is not local FXS port need go out through FXO port, such as another IAD equipment or soft switch, you can send the called number 9 to the S+O equipment, and after hearing the secondary dial tone from FXO port, and then enter the called number out.

5.2 FXS+FXO Equipment FXO port configuration—corresponding FXS port

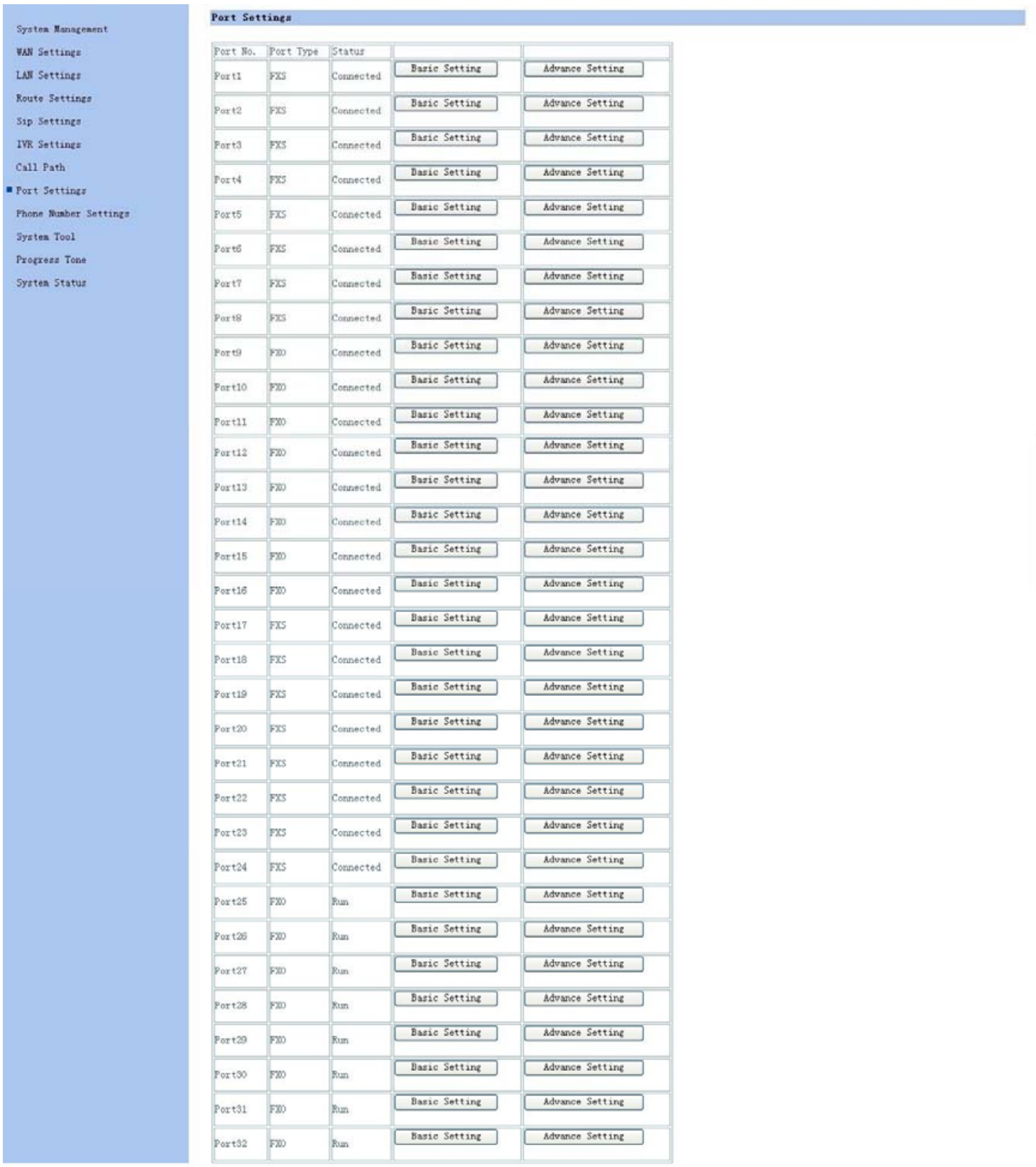
Sign in WEB configuration interface, and the detail configuration process is as follows:

1 select “Phone number setting”, view the number in current FXS port, as following figure(figure5-12):

Name	Port Type	Status		
6001	FXS	Activated	Edit	Delete
6002	FXS	Activated	Edit	Delete
6003	FXS	Activated	Edit	Delete
6004	FXS	Activated	Edit	Delete
6005	FXS	Activated	Edit	Delete
6006	FXS	Activated	Edit	Delete
6007	FXS	Activated	Edit	Delete
6008	FXS	Activated	Edit	Delete
6009	FXO	Activated	Edit	Delete
6010	FXO	Activated	Edit	Delete
6011	FXO	Activated	Edit	Delete
6012	FXO	Activated	Edit	Delete
6013	FXO	Activated	Edit	Delete
6014	FXO	Activated	Edit	Delete
6015	FXO	Activated	Edit	Delete
6016	FXO	Activated	Edit	Delete
6017	FXS	Activated	Edit	Delete
6018	FXS	Activated	Edit	Delete
6019	FXS	Activated	Edit	Delete
6020	FXS	Activated	Edit	Delete
6021	FXS	Activated	Edit	Delete
6022	FXS	Activated	Edit	Delete
6023	FXS	Activated	Edit	Delete
6024	FXS	Activated	Edit	Delete
6025	FXO	Activated	Edit	Delete
6026	FXO	Activated	Edit	Delete
6027	FXO	Activated	Edit	Delete
6028	FXO	Activated	Edit	Delete
6029	FXO	Activated	Edit	Delete
6030	FXO	Activated	Edit	Delete
6031	FXO	Activated	Edit	Delete
6032	FXO	Activated	Edit	Delete

Figure 5-12 view the number

2.According to the above figure, the number in FXS port is 6001-6008,6017-6024。Enter“port bulk configuration”，you can see the port type of 1-8 port are FXS port, 9-16 port are FXS port, 17-24 port are FXS port, 25-32 port are FXO port, as following figure(Figure5-13):



Port No.	Port Type	Status		
Port1	FXS	Connected	Basic Setting	Advance Setting
Port2	FXS	Connected	Basic Setting	Advance Setting
Port3	FXS	Connected	Basic Setting	Advance Setting
Port4	FXS	Connected	Basic Setting	Advance Setting
Port5	FXS	Connected	Basic Setting	Advance Setting
Port6	FXS	Connected	Basic Setting	Advance Setting
Port7	FXS	Connected	Basic Setting	Advance Setting
Port8	FXS	Connected	Basic Setting	Advance Setting
Port9	FXO	Connected	Basic Setting	Advance Setting
Port10	FXO	Connected	Basic Setting	Advance Setting
Port11	FXO	Connected	Basic Setting	Advance Setting
Port12	FXO	Connected	Basic Setting	Advance Setting
Port13	FXO	Connected	Basic Setting	Advance Setting
Port14	FXO	Connected	Basic Setting	Advance Setting
Port15	FXO	Connected	Basic Setting	Advance Setting
Port16	FXO	Connected	Basic Setting	Advance Setting
Port17	FXS	Connected	Basic Setting	Advance Setting
Port18	FXS	Connected	Basic Setting	Advance Setting
Port19	FXS	Connected	Basic Setting	Advance Setting
Port20	FXS	Connected	Basic Setting	Advance Setting
Port21	FXS	Connected	Basic Setting	Advance Setting
Port22	FXS	Connected	Basic Setting	Advance Setting
Port23	FXS	Connected	Basic Setting	Advance Setting
Port24	FXS	Connected	Basic Setting	Advance Setting
Port25	FXO	Run	Basic Setting	Advance Setting
Port26	FXO	Run	Basic Setting	Advance Setting
Port27	FXO	Run	Basic Setting	Advance Setting
Port28	FXO	Run	Basic Setting	Advance Setting
Port29	FXO	Run	Basic Setting	Advance Setting
Port30	FXO	Run	Basic Setting	Advance Setting
Port31	FXO	Run	Basic Setting	Advance Setting
Port32	FXO	Run	Basic Setting	Advance Setting

Figure 5-13 Port setting

3. click "basic setting" button on port 9, select "turn on" for hotline, and write "6001" in hotline No. field (6001 is the tel. number configured in port 1), select 0 in the pull down list in hotline delay field.

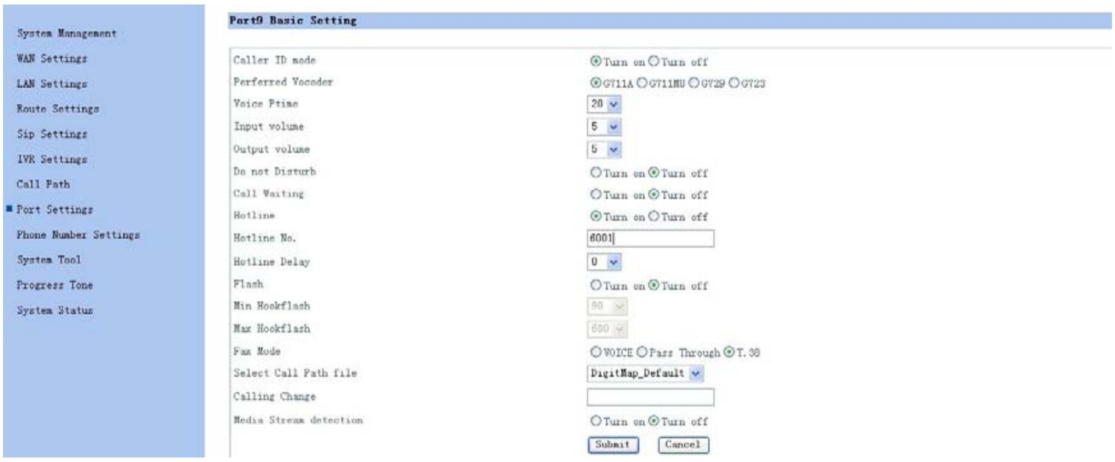


Figure 5-14 Port configuration

4.And so on，select “6002” as the hotline number for port 10, 6003 as hotline number for Port 11, 6017 as hotline number for port 25, 6024 as hotline number for port 32.

5 Enter “call path” configuration interface, edit‘Digitmap_Default’item, add a call rule named “6xxx”, route OP is 127.0.0.1 (**Terminal loopback)**，as following figure(Figure 5-15):

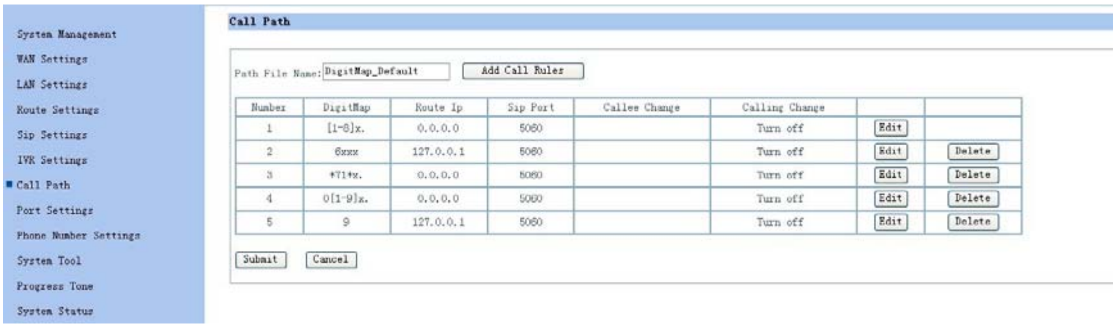


Figure 5-15 Edit call path

6. After the above configuration is finished, when dialing the PSTN number connected to FXO port through the PSTN number, the S port telephone in corresponding hotline will ring, and after off-hook in S port, normal talk with the telephone in PSTN side is ok.

Scene Summary:

This scene described the application when PSTN->FXO->FXS, the S+O port equipment through local FXO route to local FXS

The expanded application for this scene:: If you need route the call from local FXO port to other IAD equipment or soft switch platform, in hotline mode, you only need set the route IPof corresponding call rule of the hotline as the IP address of other IAD equipment, and the other IAD equipment or soft switch platform also need be able to receive the point to point calls from this S+O port equipment.

